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(54) Hybrid perceptual audio coding.

(57) A hybrid coding technique for high quality coding of audio signals, using a subband filtering technique further refined to achieve a large number of subbands. Noise masking thresholds for subbands are then determined using a new tonality measure applicable to individual frequency bands or single frequencies. Based on the thresholds so determined, input signals are coded to achieve high quality at reduced bit rates.

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HYBRID PERCEPTUAL AUDIO CODING

Field of the Invention

The present invention relates to coding of time varying signals, such as audio signals representing voice or music information.

Background of the Invention

In recent years several advanced bit rate reduction algorithms for high quality digital audio have been proposed (see e.g., Schroeder, E.F. & Voessing, W.: "High quality digital audio encoding with 3.0 bits/sample using adaptive transform coding," 80th. AES-convention Montreaux 1986, Preprint 2321 (B2); Theile, G. & Link, M. & Stoll, G.: "Low bit rate coding of high quality audio signals," 82nd AES convention, London 1987, Preprint 2432 (C-1); Brandenburg, K.: "OCF - A new coding algorithm for high quality sound signals," in Proc. of the 1987 Int. Conf. on Acoust., Speech and Signal Proc. ICASSP 1987, pp. 141-144; and Johnston, J.D.: "Transform Coding of Audio Signals Using Perceptual Noise Criteria," IEEE Journal on Selected Areas in Communications, Vol. 6 (1988), pp. 314-323). Nearly transparent quality can be achieved at bit rates down to 64 kbi/sec. using frequency domain approaches (see e.g., Brandenburg, K. & Seitzer, D.: "OCF: Coding High Quality Audio with Data Rates of 64 kbi/sec," 85th AES convention, Los Angeles 1988; Johnston, J.D.: "Perceptual Transform Coding of Wideband Stereo signals," pp. 1993-1996, ICASSP 1989; and Theile, G. & Stoll, G. & Link, M.: "Low bit-rate coding of high quality audio signal. An introduction to the MASCAM system," EBU Review- Technical, No. 230 (August 1988), pp. 71-94).

FIG. 1 shows the basic block diagram common to all perceptual frequency domain coders. A filterband 101 is used to decompose the input signal into subsampled spectral components. The subsampled spectral components are then used to calculate an estimate of the actual (time dependent)) masking threshold in block 102 using rules known from psychoacoustics (see e.g., Zwicker, E.: "Psychoakustik" (in German), Berlin Heidelberg New York 1982; Hellman, R. P.: "Asymmetry of masking between noise and tone, Perception and Pyschophysics," Vol. 11, pp. 241-246, 1972; and Scharf, B: Chapter 5 of Foundations of Modern Auditory Theory, New York, Academic Press, 1970). The spectral components are then quantized and coded in block 103 with the aim of keeping the noise, which is introduced by quantizing, below the masking threshold. Depending on the algorithm this step is done in very different ways, from simple block companding to analysis by synthesis systems using additional noiseless compression.

Finally, a multiplexer 104 is used to assemble the bitstream, which typically consists of the quantized and coded spectral coefficients and some side information, e. g. bit allocation information.

There are two filterbank designs commonly used in the above arrangement. One type is the so-called tree-structured filterbank (see e.g. QMF filterbank; descrided in Jayant, N. S. & Noll, P.: Digital Coding of Waveforms: Principles and Applications to Speech and Video, Englewood Cliffs 1984) which are designed with the filter bandwidth of the individual bands set according to the critical bands as known from psychoacoustics. Also known are those filter banks used in transform coders (see e.g., Jayant, N. S. & Noll, P.: above, and Zelinski, R. & Noll, P., "Adaptive Transform Coding of Speech Signals," IEEE Trans. on Acoustics, Speech and Signal Processing, ASSP-25 (1977), pp. 299-309) which use a windowed transform to implement a filter bank with equal bandwidth filters with low computational complexity. Transform coders typically calculate 128 to 1024 spectral components, which also can be grouped by critical bands.

The basic problem of the design of an analysis/synthesis system for use in high quality digital audio coding is the trade-off between time domain and frequency domain behavior: If more spectral components are used, the masking functions can be estimated with better accuracy. In addition, a higher decorrelation of the spectral components, and therefore a higher coding gain, can be achieved. On the other hand, a higher spectral resolution necessitates less time resolution, which leads to problems with preechoes (see e.g., Vaupelt, Th.: "Ein Kompander zur Unterdrueckung von hoerbaren Stoerungen bei dynamischen Signalpassagen fuer ein Transfor- mationscodierungsverfahren fuer qualitative hochwertige Audiosignale (MSC)", (in German), ITG Fachbericht 106, pp. 209-216; and Brandenburg, K.: "High quality sound coding at 2.5 bi(sample," 84th AES Convention, Paris 1988, Preprint 2582) and longer processing delay.

Summary of the Invention

The present invention provides structure and methods which seek to overcome the limitations of the prior art through a closer match to the processing of audio signals by the human ear. More specifically, the present

invention models the ear as a filterbank, but with differing time and frequency resolution at different frequencies. Thus the present invention provides an analysis framework that achieves a better fit to the human ear.

The hybrid coder of the present invention, in typical embodiment, uses a quadrature mirror filter to perform an initial separation of input audio signals into appropriate frequency bands. This filtered output is again filtered using a windowed transform to achieve the effect of a computationally effective filter bank with many channels.

Masking thresholds for the filtered signals are then determined using a "superblock" technique. As in earlier work by the present inventors, a "tonality" measure is used in actually developing appropriate masking thresholds. In the present invention, however, an improved tonality measure that is local to critical bands, or even a single spectral line, is used. Advantageously, well known OCF coding and quantization techniques are then used to further process the perceptually coded signals for transmission or storage.

Brief Description of the Drawing

FIG. 1 shows a general block diagram of a perceptual coder.

FIG. 2 shows a basic analysis system used in the hybrid coder of the present invention in the context of a system of the type of the type shown in FIG. 1.

FIG. 3 shows a time/frequency breakdown of the hybrid analysis structure of FIG. 2.

FIG. 4 shows a short time spectrum of a test signal.

FIG. 5 shows a block diagram of the iteration loops of a typical implementation of the present invention.

Detailed Description

THE NEW ANALYSIS/SYNTHESIS FILTERBANK

The hybrid coder in accordance with an illustrative embodiment of the present invention uses a hybrid QMF/Transform filterbank. FIG. 2 shows the basic analysis/synthesis system. The time domain values are first filtered by a conventional QMF-tree filterbank 201-203. This filterbank is used to get 4 channels with 3 to 12 kHz bandwidth (frequency resolution) and, accordingly, 2 to 8 sample time resolution. The QMF filterbank was chosen only because optimized filters were readily available that satisfied our design goals. It proves convenient to use 80-tap QMF filters derived from Johnston, J. D., *"A Filter Family Designed for Use in Quadrature Mirror Filter Banks,"* ICASSP 1980, pp. 291-294). This 80-tap filter is clearly an overdesign; lower computational complexity will clearly suffice.

It is well known that classical QMF-tree filterbanks do not yield "perfect reconstruction" of the input signal. However, 80 tap filter illustratively used yields near perfect reconstruction of the analysis/synthesis filter bank in the sense that the sum of the pass band ripple is below 16 bit resolution. Thus, rounding leads to perfect reconstruction.

The output signals of the QMF-tree are filtered again, this time using a windowed transform to get a computational effective filter bank 210-213 with many channels. The window used is a sine window, using 50 % overlap of the analysis blocks. Two different transforms have been used for this purpose. The first transform that may be used is a classical DFT, which calculates 65 or 129 (lowest frequencies) complex lines. In this approach the analysis- synthesis filterbank is not critically sampled. On the other hand, prediction of the complex frequency lines can be easily used to reduce the data rate further. Alternatively, a modified DCT (MDCT) as used in Brandenburg, K.: "Ein Beitrag zu den Verfahren und der Qualitaetsbewteilung fuer hochwertige Musikcodierung," (in German), Ph.D. thesis, Universitaet Erlangen-Nuernberg 1989 and described first in Princen, J. & Johnson, A., Bradley, A.: "Subband / Transform Coding Using Filter Bank Designs Based on Time Domain Aliasing Cancellation", in Proc. of the 1987 Int. conf. on Acoustics, Speech and Signal Processing ICASSP 87, pp. 2161-2164 may be used. This technique calculates 64 or 128 frequency values per subband and is critically sampled. Using this MDCT approach, only half the samples have to be quantized and encoded as compared to the DFT solution.

The combined filterbank has a frequency resolution of 23.4 Hz at low frequencies and 187.5 Hz at high frequencies, with a corresponding difference in time resolution. While the time resolution is illustratively quantized to powers of 2, advances in the analysis/synthesis method will provide more range in time/frequency resolution as well as less quantization. Depending on the frequency band, the characteristics of the filter bank are similar to an MDCT filterbank of block length 1024 at low frequencies and 128 at high frequencies. Thus, the frequency resolution at low frequencies is sufficient for the perceptual model, and the time resolution at high frequencies is short enough for pre-echo control without additional algorithmic accommodation. Table 1 shows time and frequency resolution values for the combined filter bank used in the hybrid coder.

Lower bound in Frequency Hz	Upper bound in Frequency Hz	Frequency Resolution Hz	Time Resolution samples	Time Resolution mS
0.0	3000.	23.4	1024	21.3
3000.	6000.	46.8	512	10.7
6000.	12000.	93.6	256	5.3
12000	24000	187.2	128	2.7

Table 1: Time and frequency resolution of the analysis/synthesis filterbank

The masking threshold is estimated using the structure of the output signal of the filterbank. The computation is done for "superblocks" containing eight "time slices" corresponding to the number of high-frequency transforms in the low-frequency transform interval. The signal energy in the lower frequency band is distributed equally between the 8 time slices, and that of the middle frequencies distributed according to their transform rate. The "superblock" allocation is shown in FIG. 3.

Then the threshold is calculated for each of the 8 time slices using improved methods similar to those in Johnston, J.D.: "Transform Coding of Audio Signals Using Perceptual Noise Criteria," IEEE Journal on Selected Areas in Communications, Vol. 6 (1988), pp. 314-323. The threshold values for transforms spread across more than 1 time slice are then added up, to give the estimate of the masking threshold with the appropriate time resolution for the critical bands contained in each transform block. Critical band boundaries are aligned with the subband boundaries, resulting in 25 critical bands.

The actual quantizer and coder must add no more noise than indicated by the estimated masking threshold in order to code the signal transparently, according to the threshold model.

CALCULATION OF TONALITY

Different values for the masking threshold for narrow band signals have been reported in literature for tone masking noise and noise as a masker. See e.g., the Hellman and Scharf references, above. In the Johnston reference, above, the spectral flatness measure was used to calculate a global "tonality" of the short time spectrum of the signal. This tonality measure was then used to interpolate between the masking threshold formulas from Hellman and Scharf. A problem has been found with the notion of a global tonality:

Some signals, especially speech signals or an "a capella" singer (see FIG. 4), show a spectrum with "tonal" parts (low harmonics of the pitch frequencies) and "noisy" parts of considerable energy at high frequencies. The result of the measurement of a global spectral flatness measure will not show that parts of the signal are very tonal (i.e., coherent from transform block to transform block). Further, i.e., even if the tonality is estimated correctly for the sensitive (tonal) parts of such a signal, the formula previously used will lead to a very conservative masking threshold at high frequencies, thereby requiring an excessive bit rate.

Experiments with changed estimated masking thresholds and results of the different approach to the estimation of the masking threshold taken in Brandenburg, K.: "Ein Beitrag zu den Verfahren und der Qualitaetsbeurteilung fuer hochwertige Musikcodierung," (in German), Ph.D. thesis, Universitaet Erlangen-Nuemberg 1989, caused a search for a new tonality measure.

As used in one aspect of the present invention to estimate the amount of masking by a signal tonality, is modeled not as a global value, but as a characteristic local to a critical band or even a single spectral line. In the context of the illustrative hybrid coder, this local tonality is estimated by a coherence measure:

For each spectral component (= subband or transform coefficient) a coherence measure is calculated. This is done using a simple prediction, calculated in polar coordinates in the complex plane. Several predictors were tested, and the one described below was selected on the basis of performance.

Let $r(t,f)$ be the radius of the spectral value at time t and frequency f and $\phi(t,f)$ the phase value at t and f.

The predicted value of r and ϕ at time t are calculated as:

$$\hat{r}(t,f)=r(t-1,f)+(r(t-1,f)-r(t-2,f))$$

$$\text{and } \hat{\phi}(t,f)=\phi(t-1,f)+\phi(t-1,f)-\phi(t-2,f)$$

The Euclidean distance between the actual and predicted values is used to get the new tonality metric, $c(t,f)$. Then,

$$c(t,f)=\frac{dist((\hat{r}(t,f),\hat{\phi}(t,f)),(r(t,f),\phi(t,f)))}{(r(t,f)+abs(\hat{p}(t,f)))}$$

If the prediction turns out to be very good, c(t,f) will have values near zero. On the other hand, for very unpredictable (noisy) signals c(t,f) will have values of up to 1 with a mean of 0.5. This "inverse tonality" or "measure of chaos" is converted to a tonality metric by a simple log-linear operation.

$$t = \alpha \ln c + \beta$$

The new tonality metric is used to estimate the masking threshold at each spectral component in the same way as described in the Johnston paper cited above for the old tonality metric. The program in Listing 1 illustrates the processing used to form c(t,f) in the context of a 512 sample input sequence. The program of Listing 1 is written in the well-known FORTRAN programming language described, e.g., in Fx/FORTRAN Programmer's Handbook, Alliant Computer Systems Corp., 1988. The program is intended for use on general purpose computers marketed by Alliant Computer Systems Corp., but may be readily adapted for use on other general purpose or special purpose processors.

In a typical version of the hybrid coder in accordance with the present teachings, the quantization and coding scheme of the OCF (Optimum Coding in the Frequency domain), system described in Brandenburg, K. & Seitzer, D.: "OCF: Coding high Quality Audio with Data Rates of 64 kbit/sec," 85th AES convention, Los Angeles 1988, has been used. In that analysis-by-synthesis scheme the spectral components are first quantized using a nonuniform quantizer. In the inner iteration loop (rate loop) the count of bits needed to code the quantized values using an entropy code is compared to the number of available bits. Depending on the ratio of actual over available bits the quantization step size is adjusted, leading to a different number of bits needed to code the block of quantized values. The outer iteration loop (distortion control loop) compares actual quantization noise energy for each critical band with the estimated masking threshold. If the actual noise exceeds the masking threshold in some critical band, the scale of the spectral components in this critical band is adjusted to yield a lower quantization noise. FIG. 5 shows a block diagram of the iteration loops used for quantization and coding. The algorithm is described in more detail in the papers by Johnston and Brandenburg and Seitzer, as well as the Brandenburg thesis, all cited above. FIG. 5 shows the manner in which a coder such as the OCF system uses the psychoacoustic threshold and related information described above to produce the actual bitstream to be transmitted or stored. Thus, input information on input 500 is assumed to have been appropriately buffered, partitioned into convenient blocks and transformed in the manner described above. The appropriate variable resolution spectral information is also provided to block 504 which provides the psychoacoustic evaluation for weighting frequency signals in block 501 prior to quantization in block 502. The actual entropy coding is represented by block 503 in FIG. 5. Thus the information describing the spectral information of the input signals is provided on output 515. Side information describing the cycle acoustic evaluation and quantizing processes is then supplied on outputs 520 and 525. All outputs are conveniently multiplexed into a single bit stream for transmission or storage.

The Perceptual Entropy (see e.g., Johnston, James D., "Estimation of Perceptual Entropy Using Noise Masking Criteria," ICASSP '88, pp. 2524-2527) is an estimate of the information content of a piece of music relative to the capabilities of the human auditory system. It gives an estimate of the minimum bit rate necessary for total transparent coding of a piece of music using a given analysis/synthesis scheme. As introduced in this last-cited paper by Johnston, the PE is calculated from the number of quantization levels necessary to code a piece of music at the masking threshold.

Using the analysis/synthesis frame work of the hybrid coder, estimates of the PE have been calculated for different pieces of music. Table 2 lists some of the results and compares them to the PE measured using other analysis/synthesis systems. It can be seen that the hybrid coder performs well compared to the older results.

music (type)	Old PE (bits/sample)	New PE (bits/sample)
organ	.24	.48
suzanne vega	.69	.54
castanets	.73	.52

Table 2: Results of PE measurements

Using the quantization/coding scheme of OCF as described above, typical results for the hybrid coder have been gathered. The bit rate used was 64 kbit/sec. per channel and the basic block length was 1024 time domain samples. The MDCT was used to compute the output of the combined filter bank from the QMF tree. The sampling rate of the test pieces was 48 kHz. The signals were coded with a bandwidth of up to 20 kHz. Out of the 1362 bits available for each block at 64 kb/s, 226 bits were used to code side information.

A second generation perceptual coder using enhanced time/frequency resolution has been described. A tonality metric, calculated on a frequency by frequency basis, is combined with the calculation of the coder's noise threshold at each frequency in order to provide a greatly improved threshold value. The present invention thus provides performance that compares favorably with known coding of high quality digital audio at low bit rates.

A decoder in accordance with the above teaching can be constructed by using the approach described above. Because of the enhanced time/frequency resolution provided by the present invention, corresponding enhanced processing is accomplished at a decoder.

Information used at a receiver or decoder to reconstruct the original input signal at the coder is, of course, that provided as outputs from the system represented by FIG. 5. In particular, the spectral information and side information, after demultiplexing if required, is used to reconstruct the original input signal. With the information describing the cycle acoustic evaluation and quantizing process, including global gain, quantizer step size, scaling factors, bit allocations and the like, all information necessary to reconstruct the sampled time domain signal from its frequency components is present at the receiver/decoder. Information about the non-uniform frequency and time resolution (both as a function of frequency) will also be used at the decoder. Well known digital to analog conversion will also be provided when it is required to create equivalent analog signals for reproduction of the original analog signal with high fidelity, e.g., on a loudspeaker.

LISTING 1

```
c       First startup routine
          subroutine strt()
c         sets up threshold generation tables, ithr and bval
          real freq(0:25)/0.,100.,200.,300.,400.,510.,630.,770.,
          1  920.,1080.,1270.,1480.,1720.,2000.,2320.,2700.,
          1 3150.,3700.,4400.,5300.,6400.,7700.,9500.,12000.,15500.,
          1 25000./
          common/thresh/ithr(26),bval(257),rnorm(257)
          common/absthr/abslow(257)
          common/sigs/ifirst
c         ithr(i) is bottom of crital band i.  bval is  bark index
c         of each line

          write(*,*) 'what spl will +-32000 be -> '
          read(*,*) abslev
          abslev=abslev-96.

          abslow=5224245.*5224245./exp(9.6*alog(10.))


          ifirst=0

          write(*,*) 'what is the sampling rate'
          read(*,*) rzotz

          fnyq=rzotz/2.
c         nyquest frequency of interest.


          ithr(1)=2.
          i=2
 10       ithr(i)=freq(i-1)/fnyq*256.+2.
          i=i+1
          if (freq(i-1) .lt. fnyq) goto 10
```

```
c     sets ithr to bottom of cb
      ithr(i:26)=257

c     now, set up the critical band indexing array

      bval(1)=0

c     first, figure out frequency, then ...

      do i=2,257,1

       fre=(i-1)/256.*fnyq
c      write(*,*) i,fre

c      fre is now the frequency of the line.  convert
c      it to critical band number..

        do j=0,25,1

         if ( fre .gt. freq(j) ) k=j
         end do
c        so now, k = last CB lower than fre
         rpart=fre-freq(k)
         range=freq(k+1)-freq(k)
         bval(i)=k+rpart/range
         end do

          rnorm=1

          do i=2,257,1
          tmp=0
          do j=2,257,1
           tmp=tmp+sprdngf(bval(j),bval(i))

           end do
```

```
      morm(i)=tmp
      end do

      morm=1./morm

c     do i=1,257,1
c     write(*,*) i, bval(i), 10.*alog10(morm(i))
c     end do

      call openas(0,'/usr/jj/nsrc/thrtry/freqlist',0)

      do i=2,257,1
      read(0,*) ii,db
      if ( ii .ne. i ) then
      write(*,*) 'freqlist is bad.'
      stop
      end if

      db=exp((db-abslev)/10.*alog(10.))
c     write(*,*) i,db
      abslow(i)=abslow(i)*db
      end do

      abslow(1)=1.
      write(*,*) 'lowest level is ', sqrt(abslow(45))

      return
      end


c     Threshold calculation program
        subroutine thrgen(rt,phi,thr)
        real r(257),phi(257)
        real rt(257)
        real thr(257)
        common/blnk/ or(257),ophi(257),dr(257),dphi(257)
```

```
common/blk1/othr(257)
real alpha(257),tr(257),tphi(257)
real beta(257),bcalc(257)
common/abslhr/abslow(257)

common/thresh/ithr(26),bval(257),rnorm(257)
common/sigs/ifirst


r=max(rt,.0005)
bcalc=1.

if (ifirst .eq. 0) then
or=0.
othr=1e20
ophi=0
dr=0
dphi=0
ifirst=1
end if

c       this subroutine figures out the new threshold values
c       using line-by-line measurement.

tr=or+dr
tphi=ophi+dphi

dr=r-or
dphi=phi-ophi

or=r
ophi=phi

alpha=sqrt((r*cos(phi)-tr*cos(tphi))
1 *(r*cos(phi)-tr*cos(tphi))
2 +(r*sin(phi)-tr*sin(tphi))
```

```
        3 *(r*sin(phi)-tr*sin(tphi)))
        4 / ( r + abs(tr) +1.)
        beta=alpha
c       now, beta is the unweighted tonality factor


        alpha=r*r
c       now, the energy is in each
c       line.  Must spread.

c       write(*,*) 'before spreading'

        thr=0.
        bcalc=0.
cvd$l   cncall
        do i=2,257,1

cvd$l   cncall
        do j=2,257,1

        glorch=sprdngf(bval(j),bval(i))
        thr(i)=alpha(j)*glorch+thr(i)
        bcalc(i)=alpha(j)*glorch*beta(j)+bcalc(i)
c       thr is the spread energy, bcalc is the weighted chaos
        end do
c       if (thr(i) .eq. 0 ) then
c       write(*,*) 'zero threshold,'
c       stop
c       end if
        bcalc(i)=bcalc(i)/thr(i)
        if (bcalc(i) .gt. .5) bcalc(i)=1.-bcalc(i)
c       that normalizes bcalc to 0-.5
        end do

c       write(*,*) 'after spreading'
        bcalc=max(bcalc,.05)
```

```
      bcalc=min(bcalc,.5)
c     bcalc is now the chaos metric, convert to the
c     tonality metric

      bcalc=-.43*alog(bcalc)-.299
c     now calculate DB

      bcalc=max(24.5,(15.5+bval))*bcalc+5.5*(1.-bcalc)

      bcalc=exp( (-bcalc/10.) * alog (10.) )
c     now, bcalc is actual tonality factor, for power
c     space.

      thr=thr*rnorm*bcalc
c     threshold is tonality factor times energy (with normalization)
      thr=max(thr,abslow)
      alpha=thr
      thr=min(thr,othr*2.)
      othr=alpha
c     write(*,*) 'leaving thrgen'
      return
      end

c     And, the spreading function
      function sprdngf(j,i)
      real i,j
      real sprdngf
c     this calculates the value of the spreading function for
c     the i'th bark, with the center being the j'th
c     bark
      temp1=i-j
      temp2=15.811389 +7.5*(temp1+.474)
      temp2=temp2- 17.5*sqrt(1.+ (temp1+.474)*(temp1+.474) )
      if ( temp2 .le. -100. ) then
      temp3=0.
      else
```

```
temp2=temp2/10.*alog(10.)
temp3=exp(temp2)
end if
sprdngf=temp3
return
end
```

TABLE I

Absolute Threshold File
("freqlist" for start-up routine)

1		56	3.	111	16.	166	16.	221	50.
2	27.	57	4.	112	17.	167	16.	222	50.
3	18.	58	4.	113	17.	168	16.	223	50.
4	16.	59	5.	114	17.	169	16.	224	50.
5	10.	60	5.	115	17.	170	16.	225	50.
6	9.	61	5.	116	18.	171	17.	226	50.
7	8.	62	6.	117	18.	172	17.	227	50.
8	8.	63	6.	118	18.	173	17.	228	50.
9	8.	64	6.	119	18.	174	17.	229	50.
10	8.	65	6.	120	18.	175	17.	230	50.
11	8.	66	7.	121	18.	176	17.	231	50.
12	7.	67	7.	122	18.	177	18.	232	50.
13	7.	68	7.	123	18.	178	18.	233	50.
14	7.	69	8.	124	17.	179	18.	234	60.
15	7.	70	9.	125	17.	180	18.	235	60.
16	7.	71	10.	126	16.	181	18.	236	60.
17	7.	72	10.	127	16.	182	19.	237	60.
18	7.	73	10.	128	16.	183	19.	238	60.
19	7.	74	10.	129	16.	184	19.	239	60.
20	7.	75	10.	130	15.	185	19.	240	60.
21	7.	76	10.	131	15.	186	19.	241	60.
22	7.	77	10.	132	15.	187	20.	242	60.
23	7.	78	10.	133	15.	188	21.	243	60.
24	7.	79	10.	134	14.	189	22.	244	60.
25	6.	80	10.	135	14.	190	23.	245	60.
26	5.	81	11.	136	13.	191	24.	246	60.
27	5.	82	11.	137	12.	192	25.	247	60.
28	5.	83	11.	138	12.	193	26.	248	60.
29	5.	84	11.	139	12.	194	27.	249	60.
30	5.	85	11.	140	12.	195	28.	250	60.
31	4.	86	12.	141	12.	196	29.	251	60.
32	4.	87	12.	142	12.	197	30.	252	60.
33	4.	88	12.	143	12.	198	31.	253	60.
34	4.	89	12.	144	13.	199	32	254	60.
35	4.	90	12.	145	13.	200	33.	255	60.
36	3	91	12.	146	14.	201	34.	256	60.
37	3.	92	13.	147	14.	202	35.	257	60.
38	3.	93	13.	148	14.	203	36.		
39	3.	94	13.	149	14.	204	37.		
40	2.	95	13.	150	14.	205	38.		
41	2.	96	13.	151	14.	206	39.		
42	1.	97	13.	152	14.	207	40.		
43	1.	98	14.	153	14.	208	41.		
44	1.	99	14.	154	14.	209	42.		
45	1.	100	14.	155	14.	210	43.		
46	0.	101	14.	156	15.	211	44.		
47	0.	102	15.	157	15.	212	45.		
48	0.	103	15.	158	15.	213	46.		
49	0.	104	15.	159	15.	214	47.		
50	0	105	15.	160	15.	215	48.		
51	0.	106	15.	161	15.	216	49.		
52	2.	107	16.	162	15.	217	50.		
53	2.	108	16.	163	15.	218	50.		
54	2.	109	16.	164	15.	219	50.		
55	3.	110	16.	165	15.	220	50.		

Claims

1. A method of processing an ordered time sequence of audio signals partitioned into blocks of samples, said method comprising
   determining a discrete short-time spectrum, $S(\omega_i)$, $i=1, 2,...,N$, for each of said blocks,
   determining the value of a tonality function as a function of frequency, and
   based on said tonality function, estimating the noise masking threshold for each of ω_i,
   CHARACTERIZED IN THAT
   said step of determining $S(\omega_i)$ comprises determining $S(\omega_i)$ with differing time and frequency resolution as a function of ω_i.

2. The method of claim 1 further
   CHARACTERIZED IN THAT
   said step of determining $S(\omega_i)$ comprises determining $S(\omega_i)$ with frequency and time resolution approximating that of human auditory response.

3. The method of claim 1 further
   CHARACTERIZED IN THAT
   predicting, for each block, an estimate of the values for each $S(\omega_i)$ based on the values for $S(\omega_i)$ for one or more prior blocks,
   determining for each frequency ω_i a randomness metric based on respective ones of the predicted value for $S(\omega_i)$ and the actual value for $S(\omega_i)$ for each block,
   said method further comprises the step of
   quantizing said $S(\omega_i)$ based on said noise masking threshold at respective ω_i.

4. The method of claim 3 further
   CHARACTERIZED IN THAT
   said step of predicting comprises,
   for each ω_i, forming the difference between the value of $S(\omega_i)$ for the corresponding ω_i from the two preceding blocks, and
   adding said difference to the value for $S(\omega_i)$ from the immediately preceding block.

5. The method of claim 4,
   CHARACTERIZED IN THAT
   said $S(\omega_i)$ is represented in terms of its magnitude and phase, and said difference and adding are effected separately for both magnitude and phase of $S(\omega_i)$.

6. The method of claim 3,
   CHARACTERIZED IN THAT
   said determining of said randomness metric is accomplished by calculating the euclidian distance between said estimate of $S(\omega_i)$ and said actual value for $S(\omega_i)$.

7. The method of claim 6,
   CHARACTERIZED IN THAT
   said determining of said randomness metric further comprises normalizing said euclidian distance with respect to the sum of the magnitude of said actual magnitude for $S(\omega_i)$ and the absolute value of said estimate of $S(\omega_i)$.

8. The method of claim 1,
   CHARACTERIZED IN THAT
   said estimating of the noise masking threshold at each ω_i comprises
   determining the energy of said audio signal at ω_i, and said method further comprises
   spreading said energy values at a given ω_i to one or more adjacent frequencies, thereby to generate a spread energy spectrum, and
   determining a desired noise level at each ω_i based on said tonality function and said spread energy spectrum for the respective ω_i.

9. The method of claim 8, wherein said estimating of the noise masking threshold function further comprises modifying said threshold function in response to an absolute noise masking threshold for each ω_i to form a limited threshold function.

10. The method of claim 9,
CHARACTERIZED IN THAT
said method further comprising modifying said limited threshold function to eliminate any existing pre-echoes, thereby generating an output threshold function value for each ω_i.

11. The method of any of claims 1, 8, 9 or 10, further
CHARACTERIZED IN THAT
said method further comprising the steps of
generating an estimate of the number of bits necessary to encode $S(\omega_i)$
quantizing said $S(\omega_i)$ to form quantized representations of said $S(\omega_i)$ using said estimate of the number of bits, and
providing to a medium a coded representation of said quantized values and information describing about how said quantized values were derived.

12. A method for decoding an ordered sequence of coded signals comprising first code signals representing values of the frequency components corresponding to a block of values of an audio signal and second code signals representing information about how said first signals were derived to represent said audio signal with reduced perceptual error, said method comprising
using said second signals to determine quantizing levels for said audio signal which reflect a reduced level of perceptual distortion,
reconstructing quantized values for said frequency content of said audio signal in accordance with said quantizing levels, and
transforming said reconstructed quantized spectrum to recover an estimate of the audio signal,
CHARACTERIZED IN THAT
said frequency components have variable rime and frequency resolution.

13. The method of claim 12
CHARACTERIZED IN THAT
said second signals identify the variation of said resolution as a function of frequency, and
said reconstructing comprises using said second signals to effect scaling of said quantized values.

14. The method of claim 12
CHARACTERIZED IN THAT
said reconstructing comprises applying a global gain factor based on said second signals.

15. The method of claim 12
CHARACTERIZED IN THAT
said reconstructing comprises determining quantizer step size as a function of frequency component.

16. The method of claim 12
CHARACTERIZED IN THAT
said second signals include information about the degree of coarseness of quantization as a function of frequency component.

17. The method of claim 12
CHARACTERIZED IN THAT
said second signals include information about the number of values of said audio signal that occur in each block.

18. Apparatus for processing an ordered time sequence of audio signals partitioned into blocks of samples comprising,
means for determining a discrete short-time spectrum, $S(\omega_i)$, i=1, 2,...,N, for each of said blocks,
means for determining the value of a tonality function as a function of frequency, and

means for estimating the noise masking threshold for each of ω_l, in response to said tonality function, said means for determining further comprises means for determining $S(\omega_l)$ with dithering time and frequency resolution at different values of ω_l,

19. The method of claim 1 further
CHARACTERIZED IN THAT
said means for determining $S(\omega_l)$ comprises determining $S(\omega_l)$ with frequency and time resolution approximating that of human auditory response.

20. The apparatus of claim 18
CHARACTERIZED IN THAT
said means for determining $S(\omega_l)$ comprises
means for partitioning said audio signal into a plurality of frequency subband, and
means for determining the short-time spectrum for each subband.

21. The apparatus of claim 20
CHARACTERIZED IN THAT
said means for partitioning comprises quadrature mirror filter means.

22. The apparatus of claim 21
CHARACTERIZED IN THAT
said quadrature mirror filter means comprises a tree structural array of quadrature mirror filters.

FIG. 1

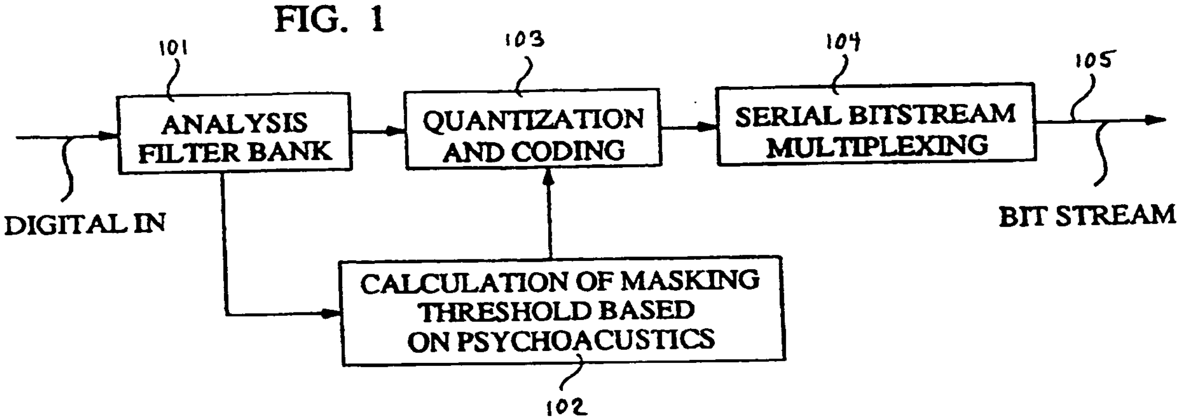

101
103
104
105
ANALYSIS FILTER BANK
QUANTIZATION AND CODING
SERIAL BITSTREAM MULTIPLEXING
DIGITAL IN
BIT STREAM
CALCULATION OF MASKING THRESHOLD BASED ON PSYCHOACUSTICS
102

FIG. 2

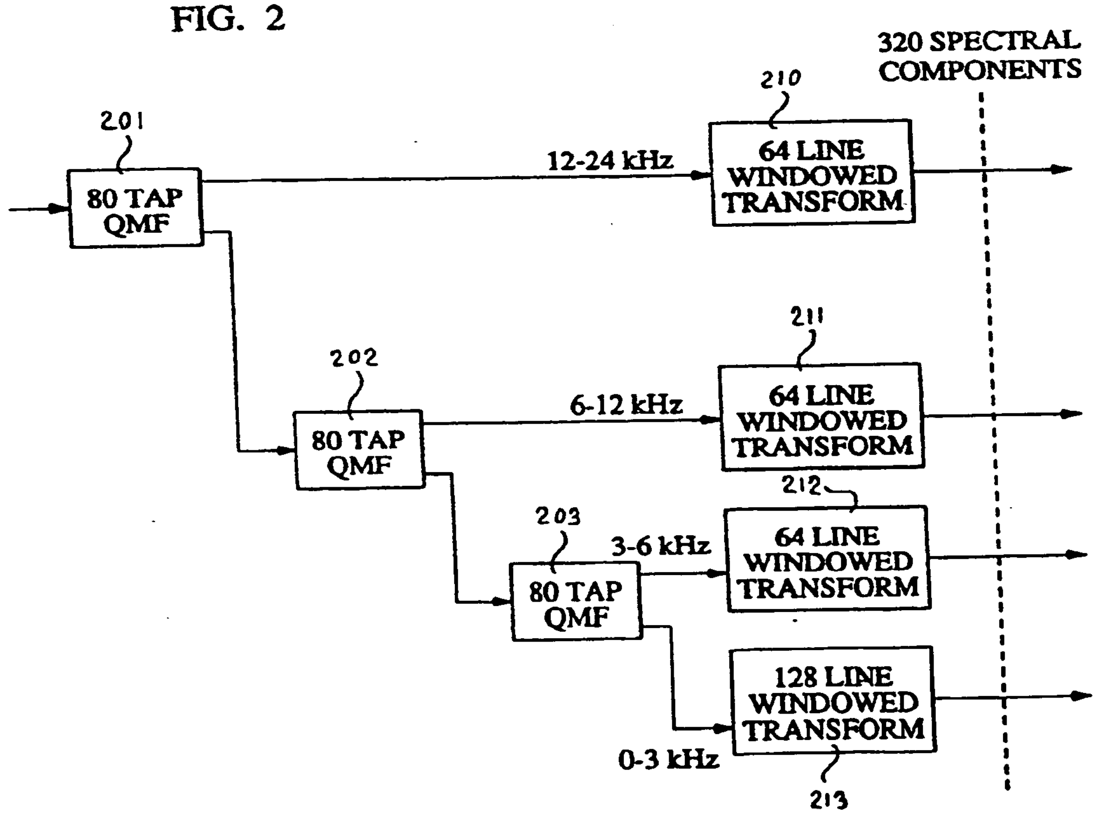

320 SPECTRAL COMPONENTS
210
201
12-24 kHz
80 TAP QMF
64 LINE WINDOWED TRANSFORM
211
202
6-12 kHz
80 TAP QMF
64 LINE WINDOWED TRANSFORM
212
203
3-6 kHz
80 TAP QMF
64 LINE WINDOWED TRANSFORM
128 LINE WINDOWED TRANSFORM
0-3 kHz
213

FIG. 3

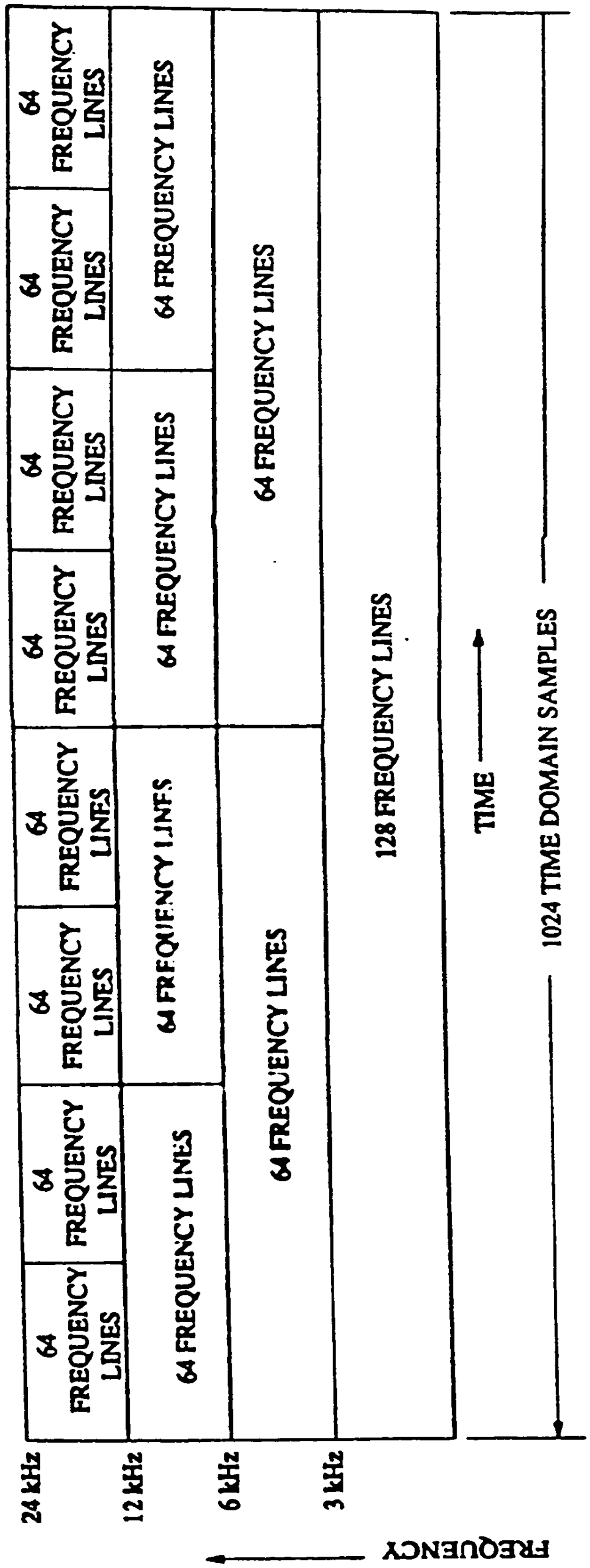

64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
64 FREQUENCY LINES
128 FREQUENCY LINES
TIME
1024 TIME DOMAIN SAMPLES
24 kHz
12 kHz
6 kHz
3 kHz
FREQUENCY

FIG. 4

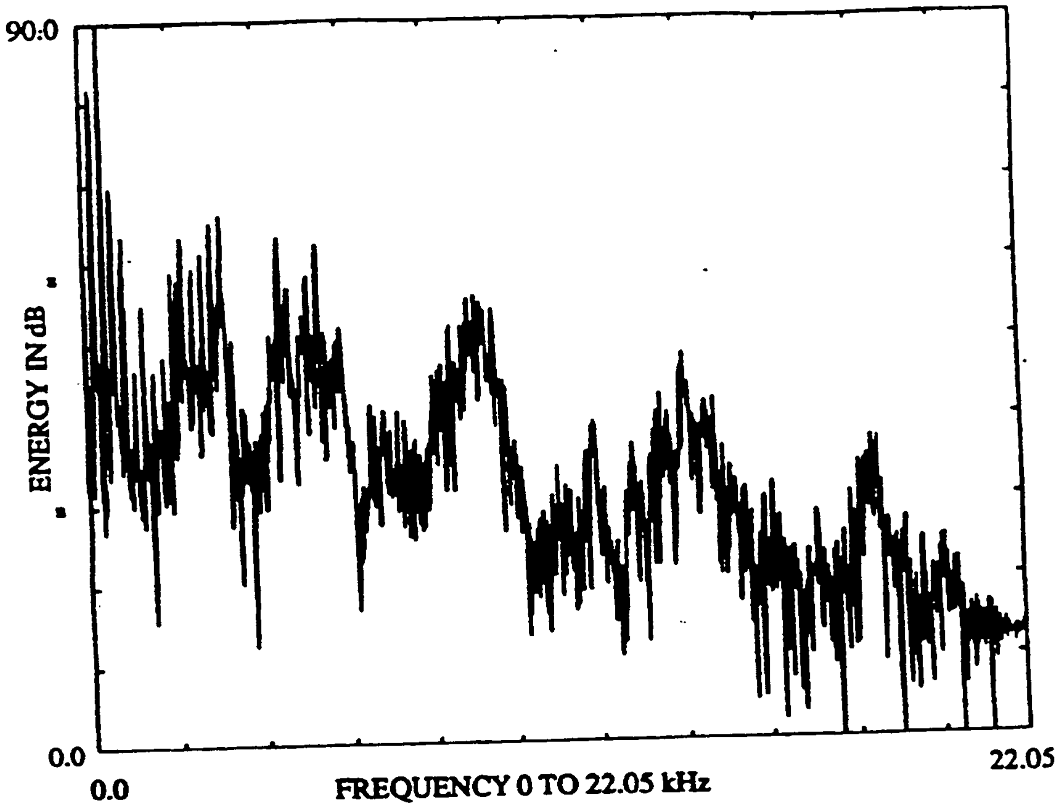

90.0
ENERGY IN dB
0.0
0.0
FREQUENCY 0 TO 22.05 kHz
22.05

FIG. 5

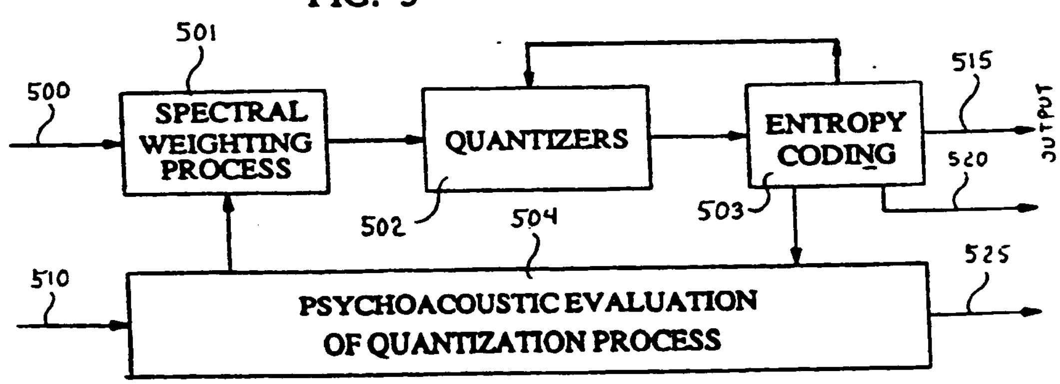

501
500
SPECTRAL WEIGHTING PROCESS
QUANTIZERS
ENTROPY CODING
515
OUTPUT
520
502
504
503
510
525
PSYCHOACOUSTIC EVALUATION OF QUANTIZATION PROCESS

FIG. 1

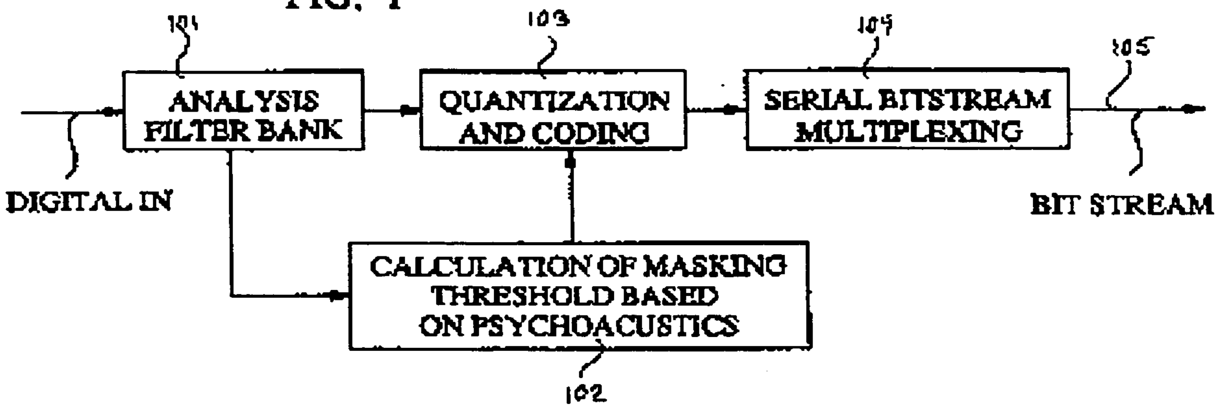

101
103
104
105
ANALYSIS FILTER BANK
QUANTIZATION AND CODING
SERIAL BITSTREAM MULTIPLEXING
DIGITAL IN
BIT STREAM
CALCULATION OF MASKING THRESHOLD BASED ON PSYCHOACUSTICS
102

FIG. 2

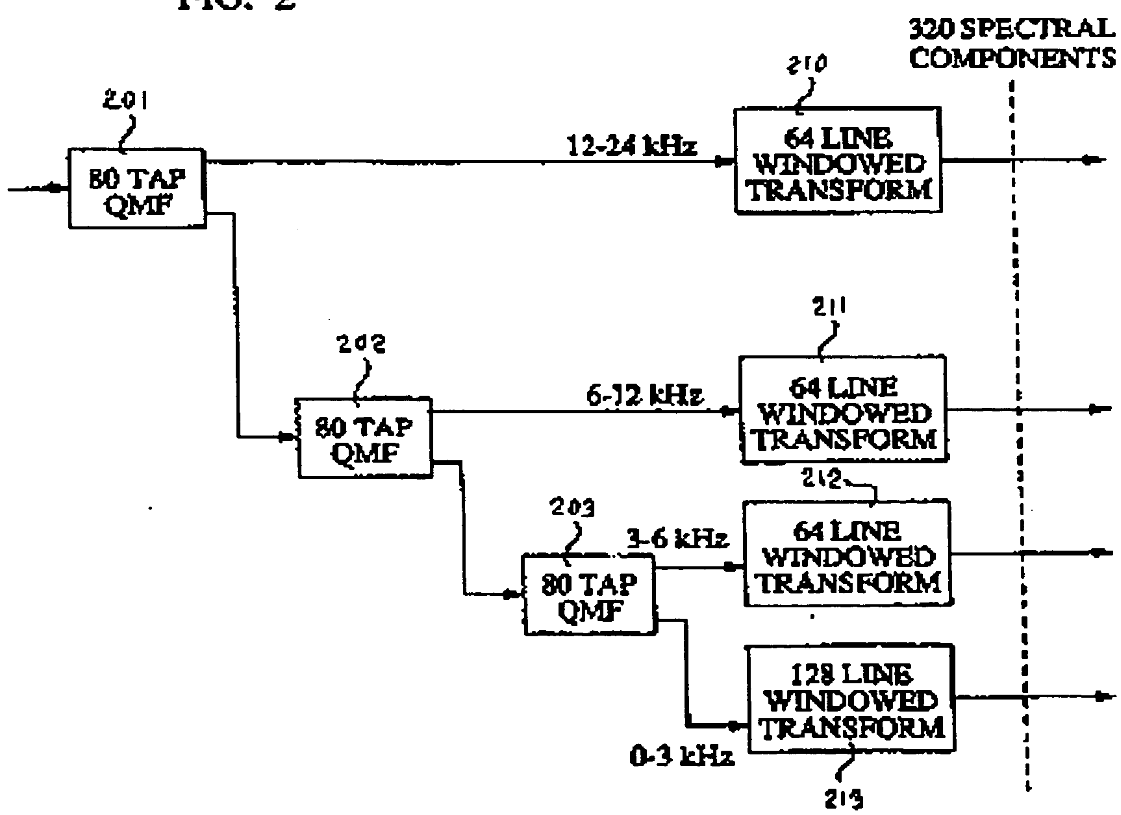

320 SPECTRAL COMPONENTS
210
201
80 TAP QMF
12-24 kHz
64 LINE WINDOWED TRANSFORM
211
202
80 TAP QMF
6-12 kHz
64 LINE WINDOWED TRANSFORM
212
203
3-6 kHz
80 TAP QMF
64 LINE WINDOWED TRANSFORM
128 LINE WINDOWED TRANSFORM
0-3 kHz
213

FIG. 3

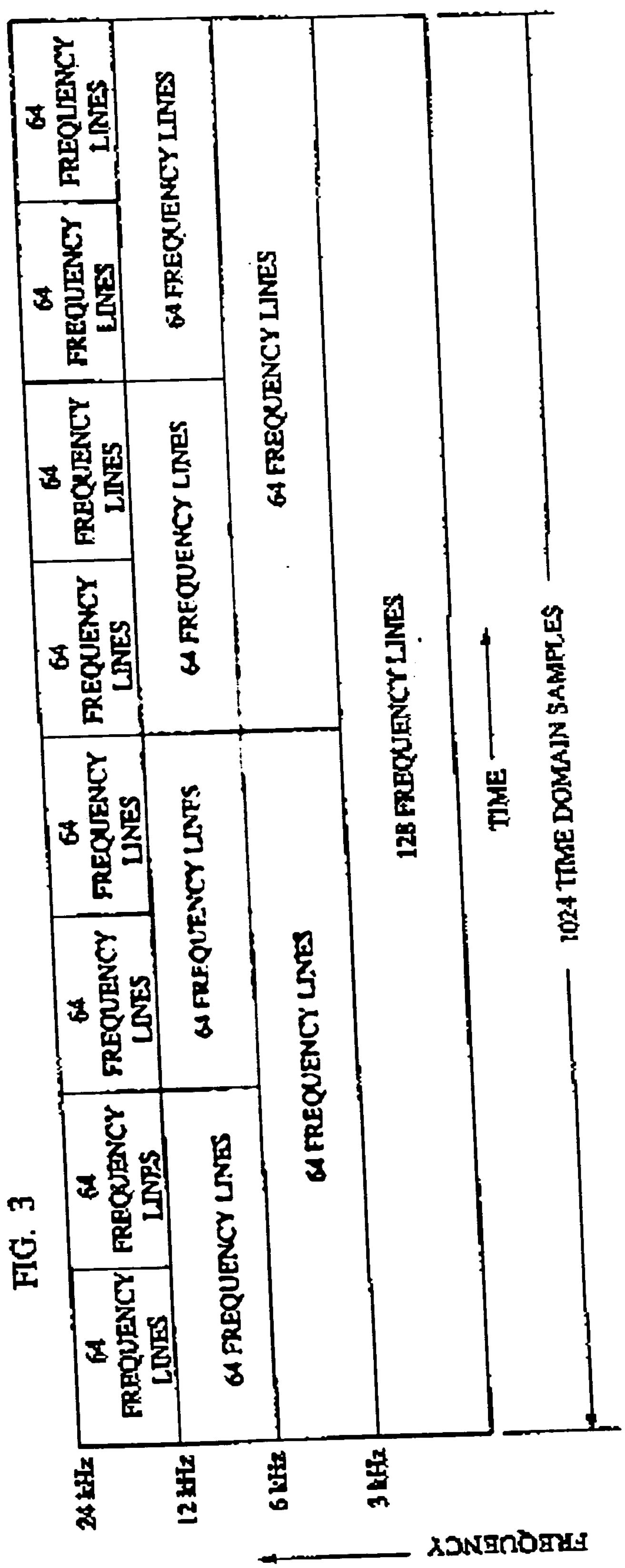

64 FREQUENCY LINES
64 FREQUENCY LINES
128 FREQUENCY LINES
24 kHz
12 kHz
6 kHz
3 kHz
FREQUENCY
TIME
1024 TIME DOMAIN SAMPLES

FIG. 4

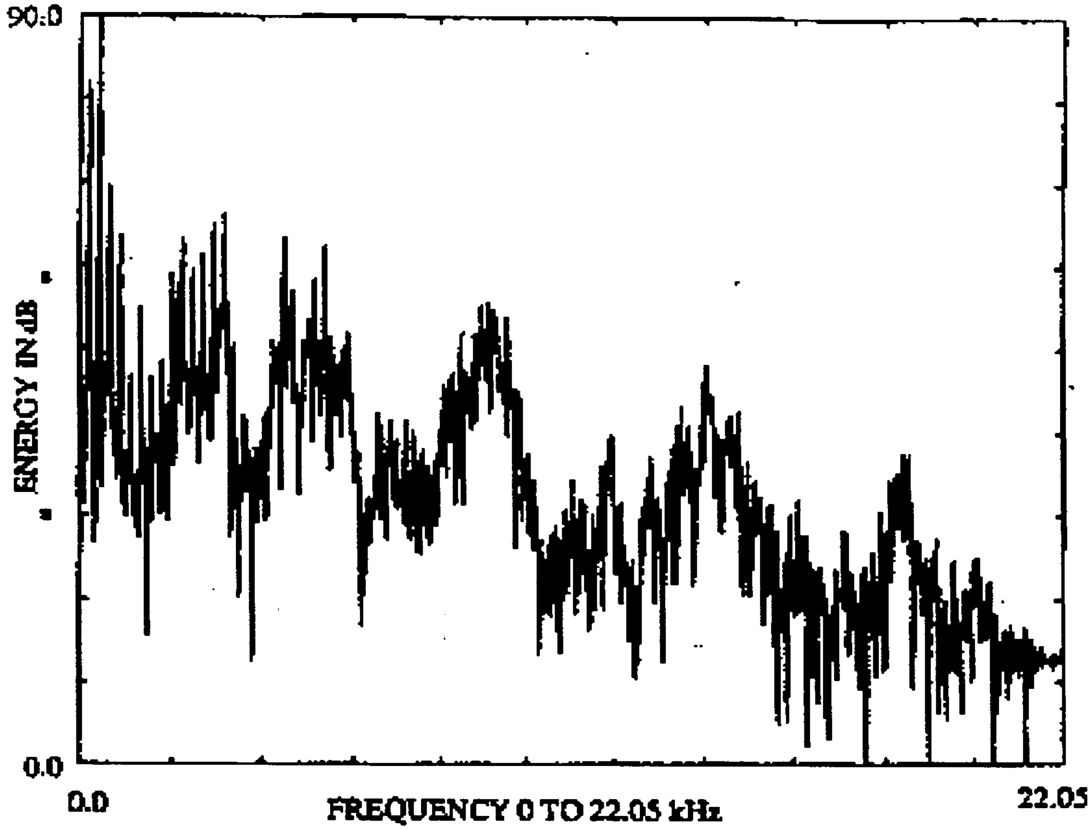
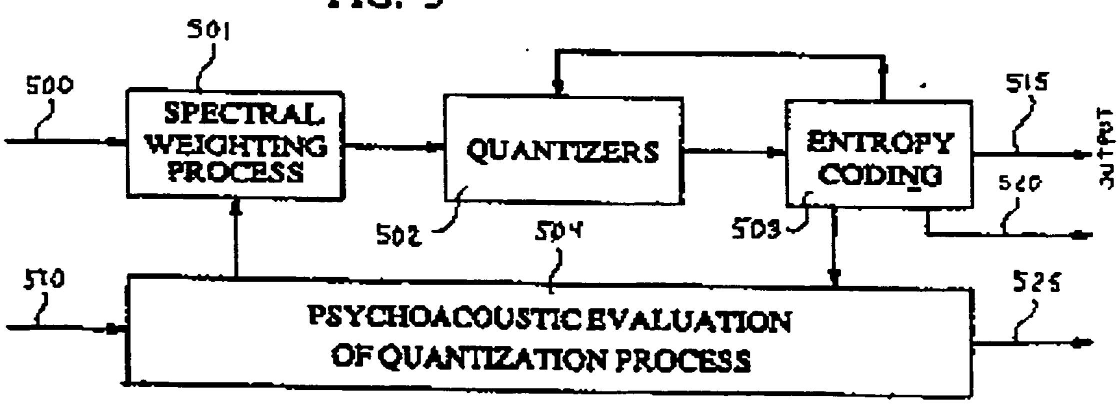

90.0
ENERGY IN dB
0.0
0.0
FREQUENCY 0 TO 22.05 kHz
22.05

FIG. 5


501
500
SPECTRAL WEIGHTING PROCESS
QUANTIZERS
ENTROPY CODING
515
OUTPUT
520
502
504
503
510
525
PSYCHOACOUSTIC EVALUATION OF QUANTIZATION PROCESS